I n t e r n a t i o n a l   T e l e c o m m u n i c a t i o n   U n i o n

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.799.3

(05/2011)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

Digital terminal equipments – Other terminal equipment

Signal processing functionality and performance of an IP-to-IP voice gateway optimized for the transport of voice and voiceband data

Recommendation ITU-T G.799.3

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Recommendation ITU-T G.799.3

Signal processing functionality and performance of an IP-to-IP voice gateway optimized for the transport of voice and voiceband data

Summary

As a consequence of the growth of IP and of IP services, different operators have an increased need to support bulk interconnection of voice and voiceband data traffic at the IP level.

Recommendation ITU-T G.799.3 defines the signal processing functionality and performance of an IP-to-IP gateway that is optimized for the transport of voice and voiceband data. Such a gateway may be used by network operators for the bulk interconnection of voice and voiceband data traffic at the IP level. This Recommendation addresses the following areas of such an IP-to-IP gateway: codec transcoding packetization time, voice performance, facsimile, voiceband data and text telephone support, support and performance of in-band signalling tones, and jitter handling. This Recommendation does not define any new protocols but, where necessary, refers to existing protocols developed within the ITU or by other standards bodies such as the IETF.

The support of tandem-free operation (TFO) and other in-band mechanisms for improving speech quality by avoiding tandem codecs will be developed in future editions of this Recommendation.

History

Edition	Recommendation	Approval	Study Group
1.0	ITU-T G.799.3	2011-05-14	16

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure, e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

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Recommendation ITU-T G.799.3

Signal processing functionality and performance of an IP-to-IP voice gateway optimized for the transport of voice and voiceband data

1 Scope

This Recommendation defines the signal processing functionality and performance of an IP-to-IP gateway that is optimized for the transport of voice and voiceband data. Such a gateway may be used by network operators for the bulk interconnection of voice and voiceband data traffic at the IP level. This is not meant to restrict an IP-to-IP gateway from being used within one provider's network.

The following areas are addressed by this Recommendation:

1) Codec transcoding
2) Packetization time
3) Voice performance
4) Facsimile, voiceband data and text telephone support and performance
5) Support and performance of in-band signalling tones
6) Jitter handling.

This Recommendation does not define any new protocols but, where necessary, refers to existing protocols developed within the ITU or by other standards development organizations.

Support of tandem-free operation (TFO) and other in-band mechanisms for improving speech quality by avoiding tandem codecs is for further study in this Recommendation.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T G.107] Recommendation ITU-T G.107 (2009), *The E-model: a computational model for use in transmission planning*.

[ITU-T G.113] Recommendation ITU-T G.113 (2007), *Transmission impairments due to speech processing*.

[ITU-T G.114] Recommendation ITU-T G.114 (2003), *One-way transmission time*.

[ITU-T G.177] Recommendation ITU-T G.177 (1999), *Transmission planning for voiceband services over hybrid Internet/PSTN connections*.

[ITU-T G.711] Recommendation ITU-T G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

[ITU-T G.722] Recommendation ITU-T G.722 (1988), *7 kHz audio-coding within 64 kbit/s*.

[ITU-T G.722.2] Recommendation ITU-T G.722.2 (2003), *Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB).*

[ITU-T G.723.1] Recommendation ITU-T G.723.1 (2006), *Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*.

[ITU-T G.726] Recommendation ITU-T G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*.

[ITU-T G.728] Recommendation ITU-T G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction*.

[ITU-T G.729] Recommendation ITU-T G.729.1 (2007), *Reduced complexity 8 kbit/s CS-ACELP speech codec*.

[ITU-T G.729.1] Recommendation ITU-T G.729.1 (2006), *G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729*.

[ITU-T G.1020] Recommendation ITU-T G.1020 (2006), *Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks*.

[ITU-T G.1050] Recommendation ITU-T G.1050 (2007), *Network model for evaluating multimedia transmission performance over Internet Protocol*.

[ITU-T H.248] Recommendation ITU-T H.248 (2000), *Gateway control protocol*.

[ITU-T H.248.1] Recommendation ITU-T H.248.1 (2005), *Gateway control protocol: Version 3*.

[ITU-T H.248.30] Recommendation ITU-T H.248.30 (2007), *Gateway control protocol: RTCP extended performance metrics packages*.

[ITU-T P.805] Recommendation ITU-T P.805 (2007), *Subjective evaluation of conversational quality*.

[ITU-T P.862] Recommendation ITU-T P.862 (2001), *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs*.

[ITU-T P.1010] Recommendation ITU-T P.1010 (2004), *Fundamental voice transmission objectives for VoIP terminals and gateways*.

[ITU-T Q.3303.2] Recommendation ITU-T Q.3303.2 (2007), *Resource control protocol No. 3 – Protocol at the interface between a Policy Decision Physical Entity (PD-PE) and a Policy Enforcement Physical Entity (PE-PE) (Rw interface):H.248 Alternative*.

[ITU-T T.38] Recommendation ITU-T T.38 (2007), *Procedures for real-time Group 3 facsimile communication over IP networks*.

[ITU-T V.150.1] Recommendation ITU-T V.150.1 (2003), *Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs*.

[ITU-T V.151] Recommendation ITU-T V.151 (2006), *Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay*.

[ITU-T V.152] Recommendation ITU-T V.152 (2005), *Procedures for supporting voice-band data over IP networks*.

[ITU-T V.153]	Recommendation ITU-T V.153 (2009), *Interworking between ITU-T T.38 and ITU-T V.152 using IP peering for real-time facsimile services*.
[ITU-T Y.1540]	Recommendation ITU-T Y.1540 (2007), *Internet protocol data communication service – IP packet transfer and availability performance parameters*.
[ITU-T Y.2012]	Recommendation ITU-T Y.2012 (2006), *Functional requirements and architecture of the NGN release 1*.
[IETF RFC 3550]	IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications*.
[IETF RFC 3611]	IETF RFC 3611 (2003), *RTP Control Protocol Extended Reports (RTCP XR)*.
[IETF RFC 4733]	IETF RFC 4733 (2006), *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*.
[TIA-1016]	TIA-1016 (2004), *Source-Controlled Variable-Rate Multimode Wideband Speech Codec (VMR-WB) Service Options 62 and 63 for Spread Spectrum Systems*.

3 Definitions

3.1 Terms defined elsewhere

3.1.1 transcoding [ITU-T V.152]: Transcoding in general is the translation from one type of encoded media format to another different media format (examples for media type "voice": ITU-T G.711 A-law to µ-law or vice versa, ITU-T G.711 codec to ITU-T G.726-40K, ITU-T G.711 to a broadband codec that operates at 256 kbit/s, etc.).

3.2 Terms defined in this Recommendation

This Recommendation defines the following term:

3.2.1 IP-to-IP voice gateway: Is a bearer level gateway (e.g., like ITU-T H.248 controlled media gateways), which provides interworking functions (or peering) between two IP domains. The term "voice" indicates that the gateway is optimized for voice, voiceband data, or 1×64 circuit-mode data services, and that the level of interworking may also be above the IP layer (e.g., similar to layer 5 (session) peering). The term "IP-to-IP" indicates that there is only unicast traffic at the gateway.

NOTE – The IP-to-IP voice gateway relates to an ITU-T H.248 controlled media gateway (MG) with two IP-based ephemeral terminations per context [ITU-T H.248]. Depending on the various scenarios, the interworking might be "media-agnostic" or "media-aware", or even "transport-protocol agnostic".

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms.

CDMA	Code Division Multiple Access
FoIP	Facsimile-over-IP
GCP	Gateway Control Protocol
GSM	Global System for Mobile communication
IP	Internet Protocol
IPDV	IP Packet Delay Variation

IPTD	IP Packet Transfer Delay
MG	Media Gateway
MGC	Media Gateway Controller
MoIP	Modem-over-IP
MOS-CQS	MOS Conversational Quality Subjective
MOS-LQO	MOS Listening Quality Objective
NAPT	Network Address And Port Translation
NGN	Next Generation Network
RR	Receiver Report (RTCP)
RTP	Real-Time Transport Protocol
RTCP	RTP Control Protocol
SDES	Source Description
SR	Sender Report (RTCP)
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
ToIP	Text-over-IP
TFO	Tandem-Free Operation
VBD	VoiceBand Data
VBDoIP	VoiceBand Data-over-IP
VoIP	Voice-over-IP
XR	eXtension Report (RTCP)

5 General description of an IP-to-IP voice gateway

5.1 IP-to-IP Voice gateway in the NGN functional architecture

Figure 1 shows the mapping of the IP-to-IP gateway to the NGN functional architecture (simplified version of Figure 3 of [ITU-T Y.2012]). In case of a distributed architecture, the figure also indicates the interface for the control of the signal processing functions located in the IP-to-IP gateway.

[ITU-T Q.3303.2] defines two IP-to-IP interworking modes: media-agnostic and media-aware. For IP-to-IP voice gateways conforming to this Recommendation, the media-aware mode should be supported.

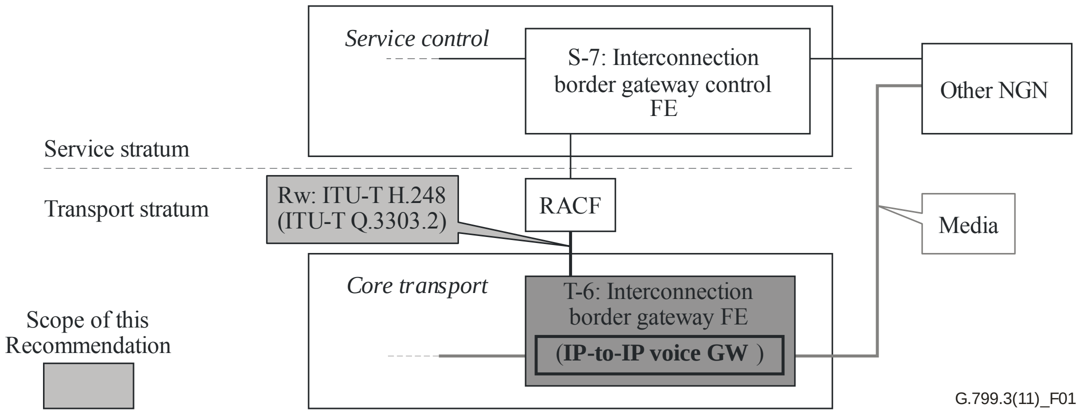


Figure 1 – IP-to-IP voice gateway in the NGN functional architecture

Figure 2 illustrates the architecture assumed in this Recommendation. An IP-to-IP voice gateway that is located at the boundary between two IP transport networks manages a set of signal processing functions at the transport plane under the control of a media gateway controller (MGC). The IP-to-IP gateway is controlled by the MGC via a gateway control protocol (GCP); see clause 5.2 and [ITU-T H.248.1].

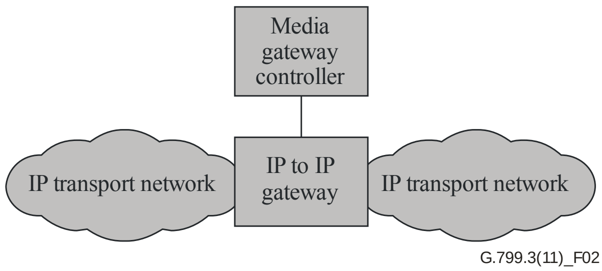


Figure 2 – IP-to-IP voice gateway

If two interconnected networks belong to different administrative domains, it is expected that each domain will have its own IP-to-IP voice gateway and associated controller (i.e., one IP-to-IP voice gateway at each side of the interconnection point), as illustrated in Figure 3.

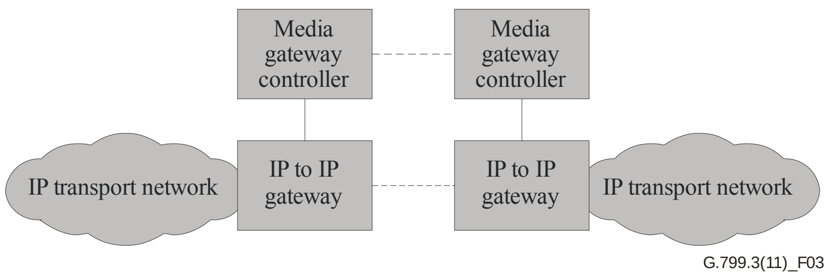


Figure 3 – Connection of two IP networks using IP-to-IP voice gateways

5.2 ITU-T H.248 profile

The ITU-T H.248 *Rw* profile, according to [ITU-T Q.3303.2], or equivalent ITU-T H.248 profile, should be used for control of IP-to-IP voice gateways defined in this Recommendation.

6 Functionality

The IP-to-IP gateway interconnects two IP networks (e.g., belonging to different operators). This means allowing the authorized media streams into the network by opening or closing firewall pinholes and providing network address and port translation (NAPT) functionality and admission control. However, these functions are outside the scope of this Recommendation.

The purpose of the IP-to-IP gateway is to provide media (payload) conversion, if necessary. This Recommendation specifies the requirements on voice and voiceband data conversion functions that may be necessary in such gateways.

The media (payload) conversion functions of the IP-to-IP gateway shall be controlled on a per-call basis.

6.1 Transcoding

The IP-to-IP gateway may perform voice transcoding. Such a need arises, e.g., if no common codecs between the endpoints can be found. If the transcoding function is performed by the IP-to-IP gateway, the following requirements apply (this does not constitute a complete list):

– At a minimum, the IP-to-IP gateway shall support ITU-T G.711 A/μ-law. The default ITU-T G.711 codec (e.g., A/μ-law) shall be selected based upon the physical location of the IP-to-IP gateway.

– An IP-to-IP gateway may optionally include codecs other than ITU-T G.711 (e.g., ITU-T G.728, ITU-T G.729, ITU-T G.723.1, and codecs commonly used in mobile networks, e.g., GSM, CDMA, TDMA). To allow for the provision of voice services with a superior quality, it is highly recommended that support for one or more wideband codec(s) such as AMR-WB [ITU-T G.722.2], VMR-WB [TIA-1016], [ITU-T G.722], or [ITU-T G.729.1] be provided by IP-to-IP gateways.

– Delay caused by voice coding should be kept to a minimum (see Annex A, and in particular clause A.3 of [ITU-T G.114], for calculation of delay figures).

If the IP-to-IP gateway performs voice transcoding, then it shall act as real-time transport protocol (RTP) translator as outlined in [IETF RFC 3550]. In this case, the IP-to-IP gateway shall modify the RTP payload type to correspond to the new codec format. In the case of transcoding between codecs with different sampling frequencies, RTP timestamps shall also be modified accordingly. In addition, the IP-to-IP gateway shall modify RTP control protocol (RTCP) packets to properly reflect the effect of transcoding in "senders octet count" field. Furthermore, the requirements of clause 6.2 apply if transcoding includes packetization time conversion.

6.2 Packetization time conversion

The IP-to-IP gateway may perform packetization time conversion. Such a need arises, e.g., if no common packet size between the endpoints can be found or in the case when transcoding is required and the codecs utilize different natural packetization times. If payload size conversion is performed by the IP-to-IP gateway, the following requirements apply.

The choice of packet size is a trade-off between transport efficiency, quality and delay, and also often by a specific supported packetization granularity of voice-over-IP (VoIP) terminal equipment. The delay associated with codec processing and packetization should be kept as short as possible. IP-to-IP gateways should adhere to [ITU-T G.177] in this regard. When multiple frames of coded speech are allocated to the same packet, packet loss concealment techniques become less effective and, as a result, may possibly lower end-to-end speech quality when packet loss is encountered.

Where codec frame size permits, an IP-to-IP gateway shall support the packetization times of 10 and 20 ms. Otherwise, an IP-to-IP gateway shall support a packetization time of one frame size (e.g., 30 ms for [ITU-T G.723.1]). These packetization times shall be supported in both transmit and receive directions. Other packetization times may be supported and negotiated under control of session or call control signalling and an MGC with the gateway control protocol (see clause 5).

Packet size shall be an integral multiple of the codec frame length. To accomplish this objective when the coding of the base [ITU-T G.729] or Annex A of [ITU-T G.729] is used, two frames per packet shall be considered as the maximum packet size.

Similarly, [ITU-T G.711] may be used with packet sizes of 10 ms (80 samples) or 20 ms (160 samples) to achieve this objective.

If the IP-to-IP gateway performs packetization time conversion, then it shall act as RTP translator as outlined in [IETF RFC 3550]. In this case, the IP-to-IP gateway shall modify RTP timestamps and sequence numbers to correspond to the modified packetization time. Losses in the incoming packet stream, if not repaired by the IP-to-IP gateway (via packet loss compensation mechanisms), shall induce corresponding gaps in the outgoing sequence numbers. The IP-to-IP gateway shall also modify RTCP packets to properly reflect packetization time conversion in "senders packet count" field, "cumulative number of packets lost" field and "extended highest sequence number received" field.

6.3 Jitter reduction (for further study)

The IP-to-IP gateway may perform packet buffering in order to reduce IP packet delay variations. Such a need arises, e.g., if the IP-to-IP gateway interconnects two IP networks with different jitter characteristics. If jitter reduction is provided by the IP-to-IP gateway, the following requirements apply.

If the IP-to-IP gateway performs jitter buffering, then it shall act as an RTP mixer as outlined in [IETF RFC 3550] (note that this does not imply that combining of RTP streams is required). Such IP-to-IP gateway shall terminate the RTCP sessions in both sides. It does not pass through sender report (RTCP) (SR) or receiver report (RTCP) (RR) packets at all; instead, it generates its own reception reports for sources in each network, and sends them out only to the same network.

The IP-to-IP gateway should forward without change the source description (SDES) information they receive from one network to the other.

The IP-to-IP gateway shall forward BYE packets.

The jitter buffer in case of voiceband data traffic shall fulfil the requirements in [ITU-T V.152], namely that it shall fix or "freeze" the jitter buffer and hence add no delay variation to the packet stream.

Example: Two interconnect IP transport domains X and Y have at a particular point in time ("during the active call and IP bearer connection") the time-averages $IPDV_X$ and $IPDV_Y$, respectively. The "jitter reduction" operation may be based on the following categories:

1) $IPDV_X \approx IPDV_Y$:

   Disabled jitter reduction function (see also clause I.1) because there would not be any significant added value.

2) $IPDV_X < IPDV_Y$ or $IPDV_X << IPDV_Y$ (see also clause I.2):

   a) **jitter elimination (or complete jitter reduction)** in the Y-to-X direction, disabled jitter reduction in the reverse direction. "Complete jitter reduction" means that the sent IP packet flow into the X domain has an IPDV value almost equal to zero; or

b) **jitter adjustment** in the Y-to-X direction, disabled jitter reduction in the reverse direction. "Jitter adjustment" means that the IP-to-IP gateway is only lowering the jitter, i.e., the IP packet flow sent into the X domain has an IPDV value similar to $IPDV_X$. The value of $IPDV_X$ could be a quasi-static parameter (i.e., provisioned in the gateway), or a dynamic parameter based on local or remote gateway measurements (e.g., via RTCP RR/XR).

6.4 Support for facsimile, voiceband data and text telephone traffic

6.4.1 Introduction

Voiceband data (VBD) traffic is always originating in a circuit-switched network (e.g., public switched telephone network (PSTN)); this may simply be an analog terminal adapter and terminal capable of VBD type calls) and may be classified as one of the following types of VBD services:

a) facsimile/modem;

b) data/modem;

c) text/modem.

The interworking of VBD services between a circuit-switched and an IP network may be achieved by packet relay modes (e.g., items 1 through 3 listed below), or by a pass-through mode such as that item 4:

1) Facsimile-over-IP (FoIP) according to [ITU-T T.38] for facsimile/modem signals;

2) Modem-over-IP (MoIP) according to [ITU-T V.150.1] for data/modem signals;

3) Text-over-IP (ToIP) according to [ITU-T V.151] for text/modem signals;

4) Voiceband data-over-IP (VBDoIP) according to [ITU-T V.152] for all type of modem-based signals.

The interworking between [ITU-T T.38] and [ITU-T V.152] using IP peering for real-time facsimile services should follow [ITU-T V.153].

6.5 Support for in-band signalling tones

6.5.1 Transfer modes for in-band telephony events, signalling and IETF RFC 4733 elements

The following basic modes of operation are possible at the bearer level of the IP-to-IP voice gateway. In Table 1, [ITU-T H.248] is used as an example of gateway control protocol.

Table 1 – Transfer modes for in-band telephony events, signalling and IETF RFC 4733 elements

No.	Mode of operation	IP-to-IP voice gateway interface
1 (Notes 1, 2)	**X** Inband **(IB)** = **X-over-G.711/TDM**	ITU-T H.248 TDM Termination
2a (Notes 2, 3)	**X** RTP Pass-Through **(PaTh)** = **X-over-VoiceCodec/RTP**	ITU-T H.248 IP Termination
2b (Note 2)	**X** RTP Packet Relay **(PaRe)** = **X-over-RFC 4733 [#]/RTP** **RTP Packet Relay submodes:** **(I) "Named Telephone Events"** (see [IETF RFC 4733]) **(II) "Telephony Tones"** (see [IETF RFC 4733], clause 4)	ITU-T H.248 IP Termination

No.	Mode of operation	IP-to-IP voice gateway interface
3 (Note 2)	**X** Out-of-Band **(OoB)** = **X-over-H.248 Package**	ITU-T H.248 Control Interface
NOTE 1 – This mode is listed for completeness, but not required for the type of gateways defined in this Recommendation. NOTE 2 – 'X' is a placeholder for in-band telephony events, signalling and IETF RFC 4733 elements as in the scope of this Recommendation. NOTE 3 – Examples are ITU-T G.711 and ITU-T G.726.		

Interworking scenarios may be required at the IP-to-IP voice gateway between transfer modes 2a, 2b and 3; see subsequent clauses for details.

6.5.2 Interworking within user plane

There may be IP domains with and without IETF RFC 4733 support and this would lead to the interworking scenario within the IP transport domain illustrated in Figure 4.

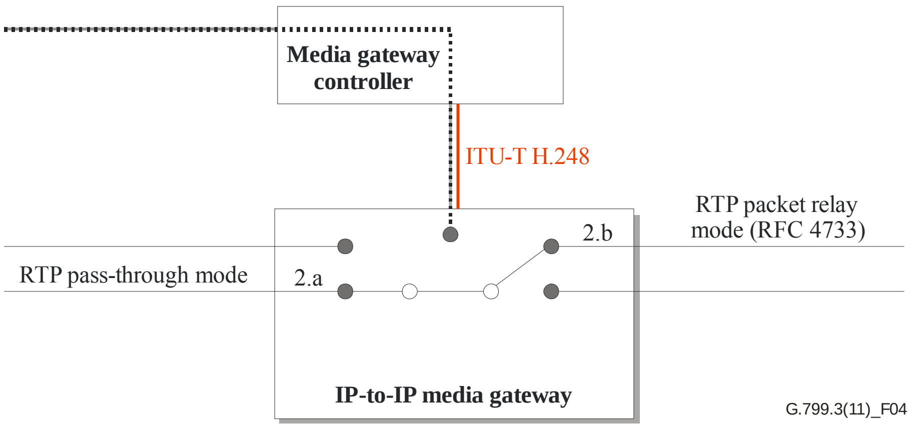


Figure 4 – Interworking between a "pass-through" and a "packet relay" domain

6.5.3 Interworking between user and control plane

Possible interworking between user and control plane interfaces of the IP-to-IP voice gateway is depicted in Figure 5.

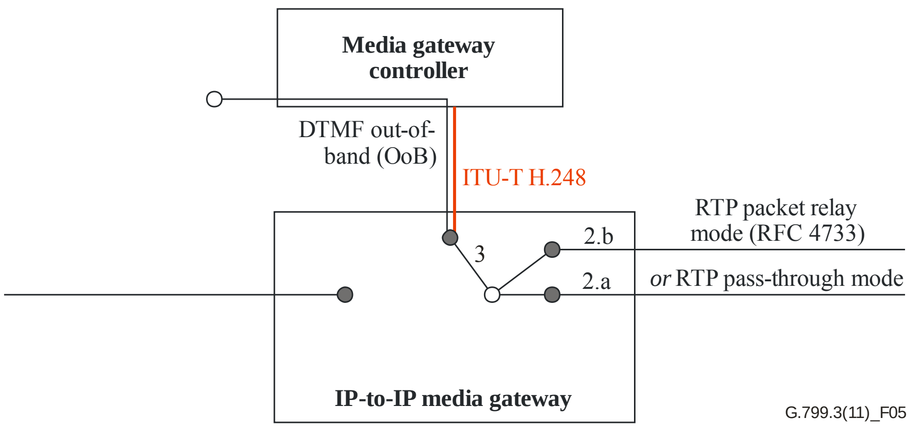


Figure 5 – Interworking between user and control plane interfaces of the IP-to-IP voice gateway

6.5.4 Non-interworking scenarios

There is no interworking in the user plane where both IP domains use the same transfer mode, see Figure 6.

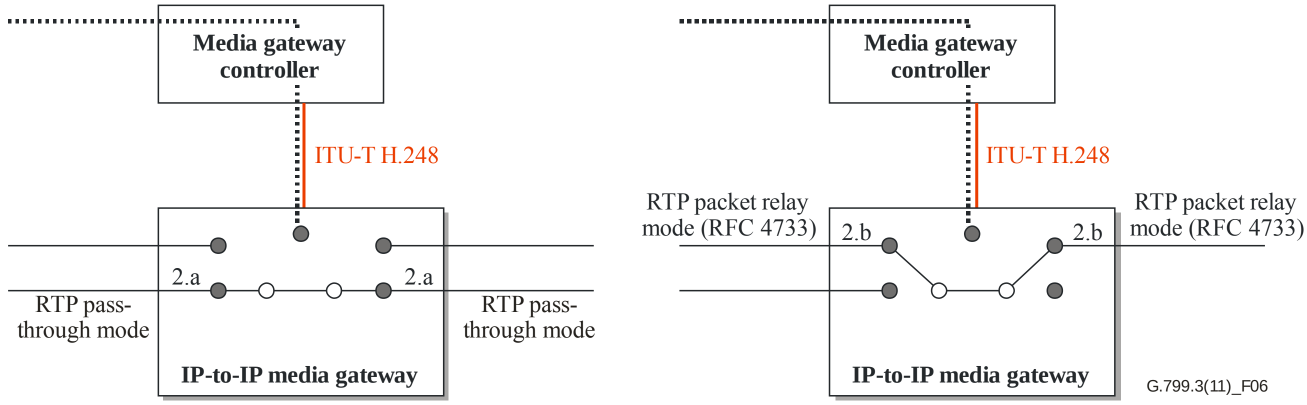
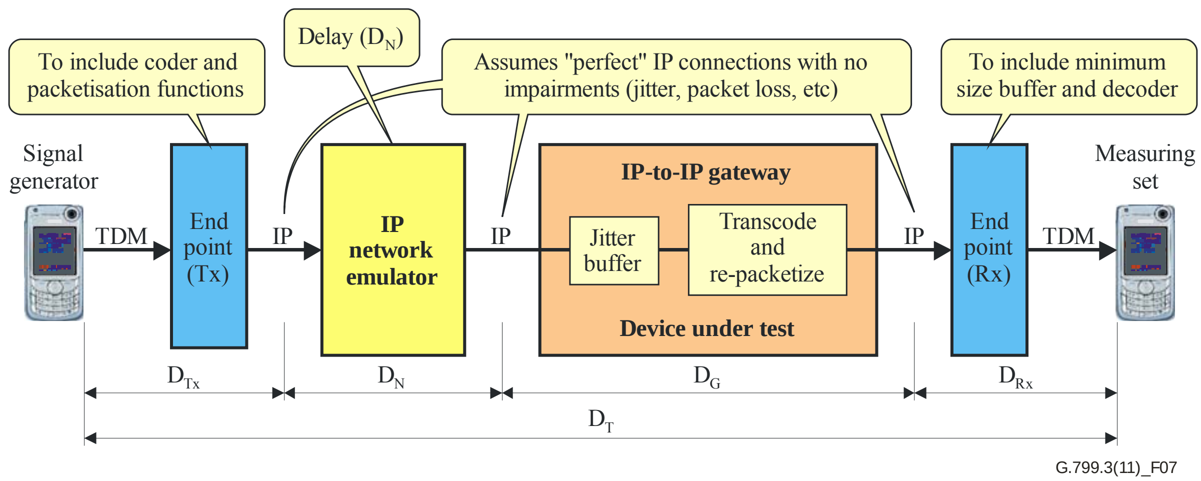


Figure 6 – Non-interworking scenarios

7 Testing and performance measurements

7.1 Testing methodology

Figure 7 illustrates a possible test set-up that may be used to measure IP packet transfer delay (IPTD) as defined in [ITU-T Y.1540] as it applies to the IP-to-IP gateway (D_G) and also end-to-end (D_T). This configuration can also be used to determine the overall end-to-end user satisfaction (e.g., the ITU-T G.107 R-factor and derived MOS), as outlined below.



NOTE 1 – Measurement shown in one direction only (left to right).

NOTE 2 – In the scenario shown, the IP network emulator only implements delay. In order to simplify the configuration, this network emulator may be omitted and an estimation of network delay be added to replace D_N.

Figure 7 – Possible test set-up for measuring IPTD parameter of IP-to-IP Gateway

The configuration should be tested in the set-up depicted in Figure 7. The IP-to-IP gateway is connected in a laboratory environment to two end-points that allow conventional TDM signals to be injected on one side of the gateway and measured at the output of the other end-point. These two end-points should be the respective gateways associated with the two networks A and B in the scenarios shown in Appendix I (note that the end-point gateway at the transmit end-point is not necessarily the same as the end-point gateway at the receive end-point). An IP network emulator is then connected in the input packet stream to the IP-to-IP gateway. This set-up allows measurement only in one direction, hence it must be repeated for measurement in the other direction.

The IP-to-IP gateway should follow the guidance outlined in [ITU-T G.1020].

The IP network emulator should comply with the requirements of [ITU-T G.1050] and should be used to introduce the fixed delay impairment D_N.

Values of D_N should be chosen to represent the typical fixed delay of the network that is going to be connected to the IP-to-IP gateway.

The end-points (Tx and Rx shown above) should comply with the requirements of [ITU-T P.1010] and are required to terminate the IP streams, giving access to the TDM signals at each end. In this respect, the transmit end-point (on the left) needs to generate packets in the relevant codec format and packet size. The receiving end-point (on the right) needs to decode the relevant codec format, remembering that there may be transcoding in the gateway. Note that the jitter buffer in this end-point should be set to a fixed value so that the value is known and can be taken into account in the determination of the IP-to-IP gateway IPTD value.

Thus, using appropriate test signals and methods (to be defined), the IPTD can be measured for the end-to-end set-up shown in Figure 6. The corresponding value for the IP-to-IP gateway under test may then be deduced, given knowledge of the performance of the individual components.

As an example, the total end-to-end delay D_T is made up of the individual delays of each component of the test set-up, i.e.,

$$D_T = D_{Tx} + D_N + \mathbf{D_G} + D_{Rx}$$

If D_T is measured, then $\mathbf{D_G}$ may be calculated since D_N, D_{Tx} and D_{Rx} are known.

The result of the end-to-end IPTD (D_T) test measurement together with the equipment impairment factor, *Ie*, for the applicable codec(s) utilized in the IP-to-IP gateway (see [ITU-T G.107] and Appendix I of [ITU-T G.113] for appropriate values) should then be compared with quality ratings shown in Figure 8 below to make sure that end-to-end user satisfaction is not degraded below acceptable limits.

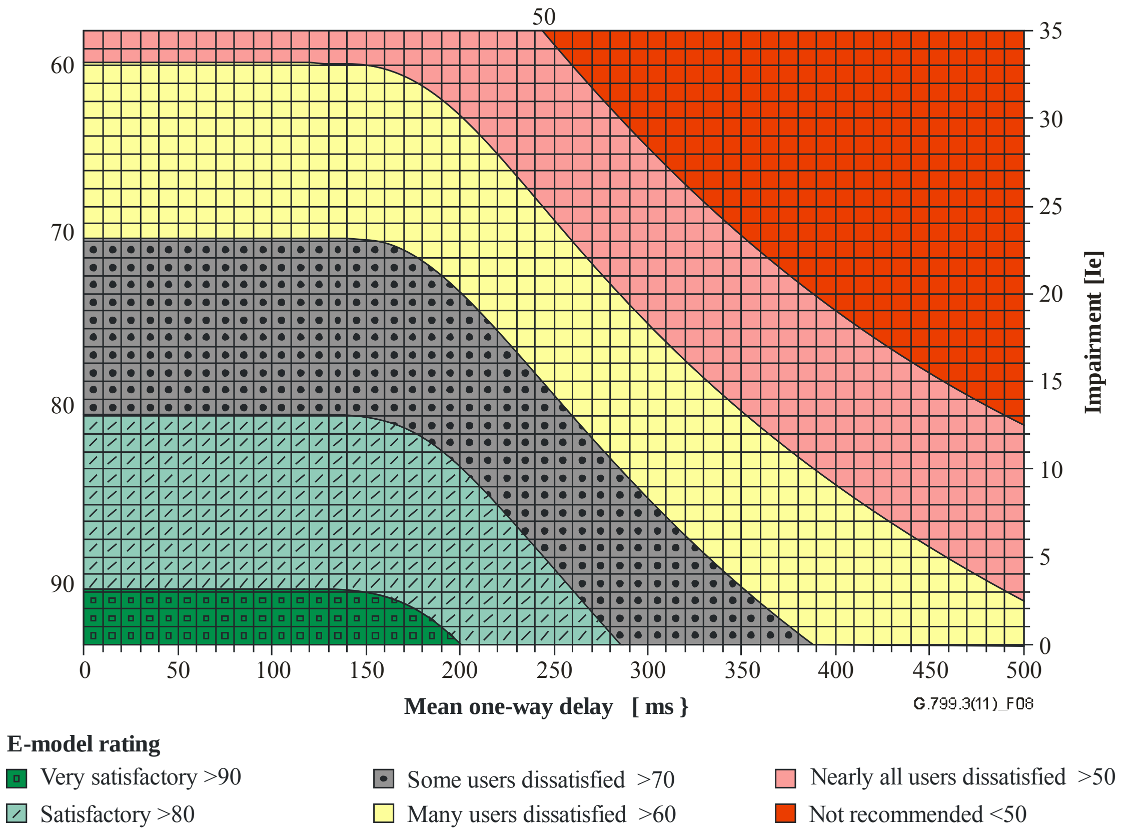


Figure 8 – User satisfaction as a function of mean one-way end-to-end delay and impairment factor (Ie)

In addition, due to the fact that the IP-to-IP gateway may contain transcoding functions, the following testing should be performed to derive end-to-end performance values for the following:

– Measure and note values for conversational quality (MOS-CQS) using ITU-T P.805 capabilities for both directions of transmission.

– Measure and note values for listening quality (MOS-LQO) using ITU-T P.862 capabilities for both directions of transmission.

– When transcoding capabilities are utilized in the IP-to-IP gateway, measure and note that level transparency is maintained between gateway input and gateway output for both directions of transmission.

8 RTCP – XR support (for further study)

If the IP-to-IP gateway is required to support characterization and reporting of voice metrics, then this support shall be provided via RTCP-XR.

The specification of RTCP-XR metrics may be found in [IETF RFC 3611].

Note that formalized parameter definitions for the [IETF RFC 3611] metrics can be found in [ITU-T G.1020].

8.1 Generation and forwarding of RTCP XR

[IETF RFC 3550] defines three types of RTP systems: the RTP end system, the mixer, and the translator. RTCP XR [IETF RFC 3611] is only defined for "RTP end systems", and an end system capable of RTCP XR expects to receive RTCP XR from its peer end-system. The IP-to-IP gateway may act either as an RTP mixer or as an RTP translator. General principles for processing of RTCP by mixers and translators are defined in [IETF RFC 3550], and these principles may be applied to processing of RTCP XR by mixers and translators. Neither [IETF RFC 3550] nor [IETF RFC 3611] addresses reporting of metrics by media gateways (such as IP-to-IP gateways) to their controllers, but the ITU-T H.248.x-series of Recommendations provides mechanisms for this reporting.

Both RTP translators and RTP mixers may make local measurements of the transport properties of incoming RTP streams and may report them to their controllers via [ITU-T H.248.30]. However, they cannot send the results of these measurements to peer RTP systems by use of RTCP packets, because RTCP XR has no provision for reporting by RTP systems which are not end systems.

8.1.1 RTCP and RTCP XR behaviour of translators

RTCP processing by a translator should be transparent to RTCP information passing between RTP end systems. This includes RTCP XR, if used. This means that an end system should be unaware that a translator is present, rather than that RTCP packets are forwarded completely unchanged. For example, where the translator performs only Layer 3 and Layer 4 processing on the RTP flow, such as modification of address and port information, it is usually necessary to perform similar Layer 3 and Layer 4 processing on the RTCP flow (such processing is outside the scope of this Recommendation). Where higher-layer processing such as transcoding occurs, RTCP and RTCP XR payload information should be transformed appropriately in accordance with clause 7.2 of [IETF RFC 3550].

Reporting of metrics by a gateway to its controller is not defined in [IETF RFC 3550]. However, where RTP end systems use RTCP XR, the translator may use the mechanisms in [ITU-T H.248.30] to report RTCP XR information sent by both end systems.

8.1.2 RTCP and RTCP XR behaviour of mixers (for further study)

[IETF RFC 3550] recommends that mixers should not forward RTCP between networks. This is clearly appropriate for transport-level metrics including those in the RTCP XR Statistics Summary block and some of those in the RTCP XR VoIP metrics block. However, it would be useful to forward some of the information in the VoIP metrics block between RTP end systems, even where an RTP mixer is present in the connection. Hence, RTCP XR processing by mixers is potentially complex and requires further study.

Appendix I

IP-to-IP gateway scenarios

(This appendix does not form an integral part of this Recommendation.)

This appendix considers different possible scenarios of an IP-to-IP gateway. The scenarios do not constitute an exhaustive list of the possible applications. The following four scenarios consider the signal processing functions associated with the media path only.

The four different scenarios given below illustrate how the IP-to-IP gateway could range from being a simple straight-through connection (Scenario 1) to a combination of jitter buffers, transcoding and re-packetization functions.

I.1 Scenario 1: Same codec, same packet size, no jitter buffer

Scenario 1 is illustrated in Figure I.1. The two networks use the same codec and packet size across the interface and the jitter handling capabilities are similar, such that the jitter buffer provided at the end-point of each network is capable of absorbing the maximum jitter introduced by the combination of both networks.

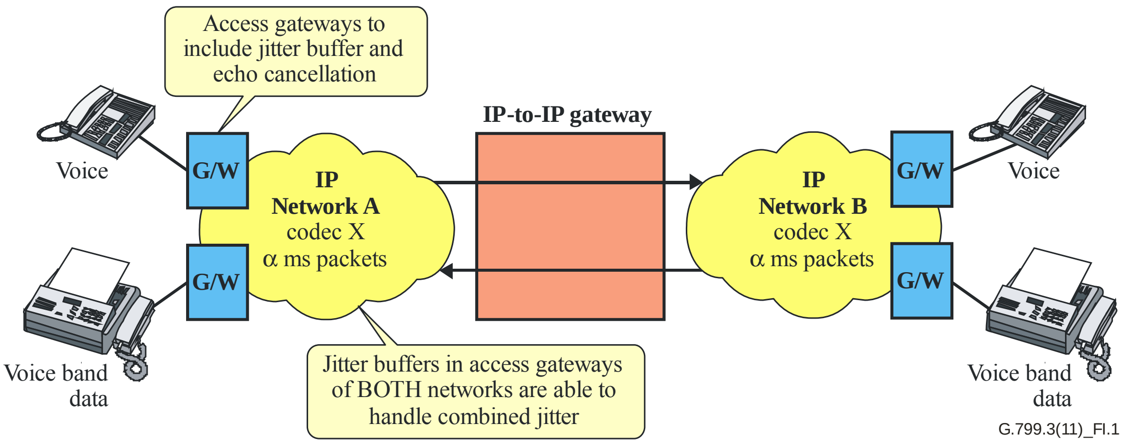


Figure I.1 – Scenario 1: Same codec, same packet size, no jitter buffer

I.2 Scenario 2: Same codec, same packet size, jitter buffer introduced to protect Network A from Network B's jitter

Scenario 2 is illustrated in Figure I.2. The two networks use the same codec and packet size across the interface but the jitter introduced by Network B is greater than that of Network A. Without a jitter buffer in the gateway, Network A's jitter buffer in the access gateway would not be able to handle the combined jitter in both networks and packet loss would occur. Conversely, Network B's access gateway jitter buffer is capable of handling the combined jitter and therefore no jitter buffer is required in the IP-to-IP gateway for traffic in the A-to-B direction.

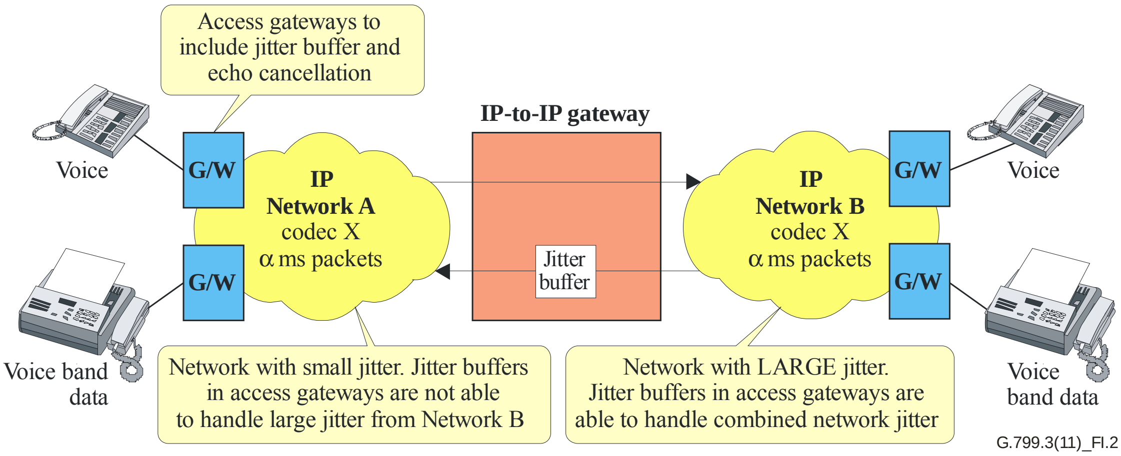


Figure I.2 – Scenario 2: Same codec, same packet size, jitter buffer in one leg

I.3 Scenario 3: Same codec, different packet size

Scenario 3 is illustrated in Figure I.3. The two networks use the same codec but the packet size across the interface is different. This may or may not require re-packetization depending on the specific codec.

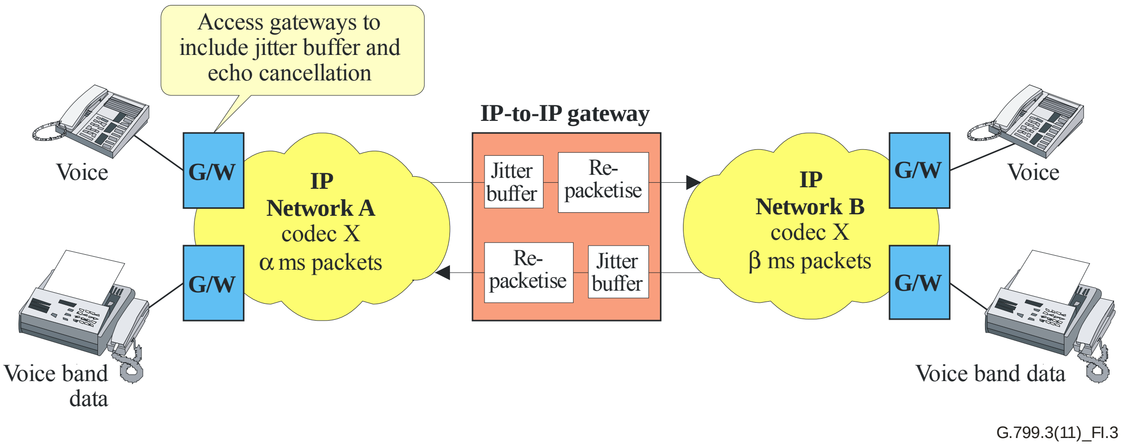


Figure I.3 – Scenario 3: Same codec, different packet size

I.4 Scenario 4: Different codec, different packet size

Scenario 4 is illustrated in Figure I.4. The two networks use different codecs and different packet sizes. Again, this may or may not require re-packetization depending on the specific codec.

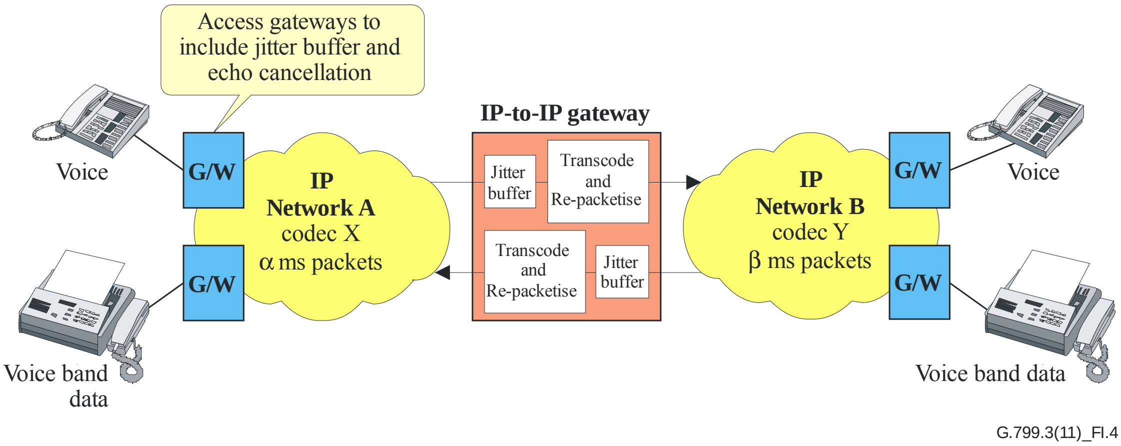


Figure I.4 – Scenario 4: Different codec, different packet size

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